Characterization and Calibration of Acoustic Radar System and Behavior Testing

Dissertation

Presented in Partial Fulfillment of the requirement for the degree of Bachelor of Science with Honor Research Distinction

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By

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2014

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Abstract

As known to all, bats have a good echolocation ability to help themselves survive in nature. Blind humans can also use echolocation to guide themselves. We can use an acoustic radar system to simulate the echolocation process. The main purpose of this research is to develop the current acoustic radar system and test its behavior to prepare for further research in comparison between bats' and human's waveforms. A stable, reliable and well calibrated system is a fundamental to all further research related to acoustic radar. In addition, by conducting this research, more details about the performance and capability of acoustic radar could be found out. Furthermore, based on the current research, because blind human are also using echolocation to help themselves, further research could be conducted with the long term goal of developing a prosthetic to help improve their perception of the environment.

Acknowledgement

Firstly, I appreciate Prof. Chris Baker and Dr. Graeme Smith's patient teaching and guidance for me in my senior year. I greatly appreciate Prof. Baker since he gave me a chance to join this research and gave me advice for career development. I have learned a lot from you about not only about the background knowledge in the relevant field, but also about how to treat research enthusiastically. I especially thank to Dr. Smith for his mentorship. He told me how to do the research step by step and helped me for Denman presentation and National Conference on Undergraduate Research. I cherish the teamwork with a MS student, Abishek Mohan, for our intimate cooperation and his valuable help. We have been closely working on this research in last academic year.

In our research, Dr. Smith helped us to acquire the necessary knowledge and background information. He patiently assigned and discuss every task to us. In this academic year, Dr. Smith guided us roughly once two weeks to teach and answer our questions.

The main purpose of my research is to develop an acoustic radar with Abishek. The radar system had some problems. Abishek and I found the reasons and corrected them. After system calibration, we tested the behavior of system by four experiments. All results of this research is from our collaboration. After conducting this research, more research will be conducted about human's and bat's waveform analysis. After system development and calibration, the acoustic radar system has become very stable and controllable by which will be able to do more research.

Our final goal is that to find out how echolocation of human and bat working with more detail, which could be helpful for man-made system.

Last but not the least, I want to thank to my parents, thanks to their support and accompany.

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Field of Study:

Electrical Engineering

Content

1. Introduction

Echolocation is an essential ability of various animals and including humans, although very few individuals are able to conduct “active echolocation”. Echolocation is the ability of sensing sound waves reflected back from objects to determine their location. The sounds wave is a type of mechanical vibration through a medium (such as air, which is the case in this research), which propagating by pressure and displacement [1]. Sound is also known as acoustics, which also includes sound, ultrasound and infrasound.

Some animals, such as bats, whale and dolphin, exploit their echolocation ability to guide themselves to avoid obstacles and to search for food. Among those animals, bat is the most typical example by using ultrasound to guide themselves [2]. Some species of bat have weak sight and others are even completely blind, thus echolocation is an indispensable ability for their survival. On the other hand, less well known example is blind humans. Some blind humans are also experts who are able to detect and discriminate a vast array of objects by using echolocation [3]. Both bats and blind people have the ability to detect different objects; however the accuracy of echolocation identification of human is only 56~86% to bat’s echolocation identification accuracy and is even a little lower than man-made system [4].

If more details could be found out and system well developed, two improved would happens. Firstly, further research about helping blind humans would be conducted by the acoustic radar system. The study in this field could have a progress. Secondly, man-made system could be improved greatly. As mentioned above, if we could know the bat's echolocation ability better, then we could definitely apply the result to a man-made system.

2. Echolocation theory

2.1 Acoustic Radar

Radar is a system which is able to transmit and receive electromagnetic waves. The main purpose for a radar is to detect the range between radar and target and to locate the target. In early age, radar can only detect the target and give its range and angle back; on the other hand, nowadays, radar can also provide information about the type of target being detected [5]. For both types of radar, the most important ability is accuracy of range detection.

In the acoustic radar system, the transmitted signal is replaced by acoustic wave. The propagation of acoustic wave is by vibration in medium. For example, air is the medium in this experiment. Acoustic signal is sent out by radar and stroked the target. When the vibration strike the target, it is be reflected in all directions including back to radar. In this experiment, we set two receivers to collect acoustic signal responses. In order to store acoustic signal response, the receivers are able to digitize the amplitude of signal vibration in air and record the digital signal.

The transmitter and two receivers are connected with a computer. LabVIEW has three main functions in this experiment. First, the transmitted signal is being set here. Before experiment, three testing signals have been set, up chirp, down chirp, up-down chirp. Second, in the front panel of LabVIEW, the detected ranges are being calculated in both channels. Third, the received signal is being stored into a CSV file with other relevant information, such as signal frequency.

2.2 Range Measurement

A radar is able to estimate the range to target by measuring the time it takes for the transmitted signal to travel to the target and back. Since the speed of propagation of sound wave in air or an EM pulse in free space can be considered a constant, v, if the round trip time of the signal is t then the range to the target will be given by

$$R_{det} = \frac{v*t}{2} \tag{1}$$

For a conventional electromagnetic radar, the velocity will be 3×10^8 ms^{-1} while for the acoustic radar used in this research the velocity is a much lower 343 ms^{-1}.

2.3 Maximum Unambiguous Range

To detect a range between the radar and a target, the radar firstly transmits a single pulse and then shuts down the transmitter and listens the corresponding response back. The limitation happens when the radar transmits the second pulse while it has not get received first echo response. The maximum unambiguous detected range in governed by the interval between two transmitted pulses, which is call pulse repetition interval (PRI). The frequency of a series of transmitted pulse is pulse repetition frequency (PRF).

$$PRI = \frac{1}{PRf} \tag{2}$$

Maximum detected range is named Maximum Unambiguous range and is given by:

$$R_{Max} = \frac{v * PRI}{2} = \frac{v}{2 * PRF} \tag{3}$$

Ambiguous range happens when the target is beyond radar maximum unambiguous range. In this case, radar receive the first response, but it has already sent out second pulse. Then, the radar system also identify the response from a second transmitted pulse automatically. The radar cannot associate which response goes with which pulse a simplified diagram for maximum Unambiguous range is shown in Figure1.

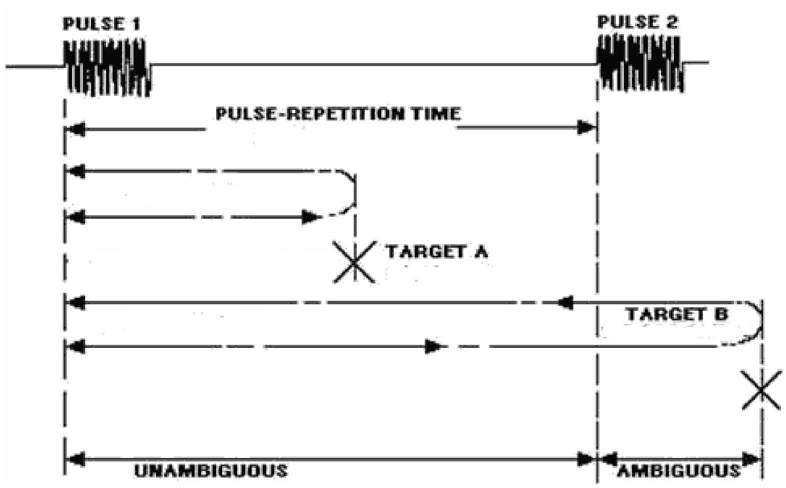


Figure 1: Maximum unambiguous theory [7]

This procedure could change measured time, t, mistakenly. In other words, when traveling time from transmitter to target, t, is larger than PRI, then time measured by radar decreases to t_{det}. The relation between real time and detected time is shown below:

$$t_{real} = i * PRI + t_{det} \tag{4}$$

Thus, when the target is beyond the unambiguous limitation, the relation between real range and detected range is shown below:

$$R_{Real} = i * R_{max} + R_{Det} \tag{5}$$

From the maximum unambiguous range equation (3), it is linearly proportional to the speed of propagation. The greater the speed of propagation the large the maximum unambiguous range becomes. As a result an electromagnetic radar will have an unambiguous range approximately one million times further than an acoustic radar for the same PRF.

On the other hand, as technology improves, people have new method to overcome this problem. The basic idea is to label one pulse and listen to that particular echo response or use other techniques, such as pulse repetition frequency, jittering and PRF switching.

2.4 Range Resolution

For pulsed radar, range resolution is the ability of a radar system to distinguish between two or more targets on the same bearing but at different ranges by using one pulse [6].

Assume a pulse width, τ, then the physical length of the pulse is shown schematically in Figure 2.

$$L = v * \tau \quad (6)$$

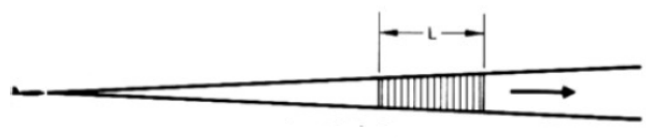


Figure 2: Physical length for transmitted wave

Assume: target A and B on a same bearing. Distance between them is D, physical pulse length is L. x is distance of echo A traveling back by the time pulse get to B. x is equal to D because transmitted pulse and echo are traveling in a same distance within a same period.

$$L - x \leq D \quad (7)$$

$$x = D \quad (8)$$

Thus,

$$D \geq \frac{L}{2} \quad (9)$$

In other words, in order to make the echo responses distinguishable, the distance between two targets must larger than the half of physical length of wave. Otherwise, as shown the two echo responses would be merged together.

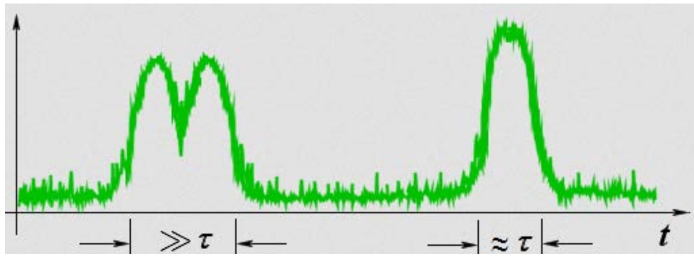


Figure 3: Ambiguous and unambiguous echo responses [7]

Thus, the minimum the range resolution is

$$R = \frac{v * \tau}{2} = \frac{v}{2 * B} \tag{10}$$

Where B is the bandwidth of the waveform and B = 1 / τ, v is speed of propagation.

For electromagnetic radar, the velocity is approximately equal to 3.0*10^8 ms^{-1}, which make minimum resolution very large unless extremely short pulses are used. Unfortunately, extremely short pulses reduce the amount of energy received back from the target and some limit detection performance. To overcome this, pulse modulation such a frequency modulation or phase coding can be used. See section 2.5 for more information of pulse modulation.

For audio radar, the speed of propagation is approximately 340 ms^{-1} leading to extremely fine range resolution. For example, if the waveform bandwidth is 10 kHz (easily achievable with acoustic hardware) the range resolution would be slightly less than 2 mm, far finer than an any operation electromagnetic radar system. In other words, it is easier to have pulse modulation bandwidth that are a bigger percentage of the center perceiving in acoustics than electromagnetics.

2.5 Pulse Compression

As mentioned above, long detection range and find range resolution are in conflict with each other. There are two commons method to solve this problem, firstly, linear frequency modulation (LFM) and, secondly, binary phase modulation (BPM).

In the linear frequency modulation method, the transmitted signal is usually down chirp or up chirp. For example, in the up chirp signal, the frequency of transmitted pulse is increased at a constant. The spectrum of up chirp is shown below.

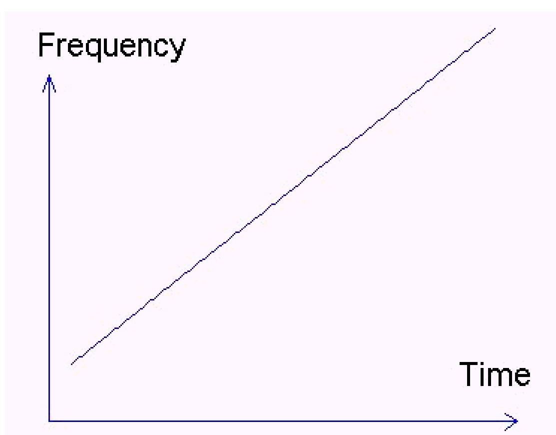


Figure 4: Spectrum of up chirp [8]

The received signals are passed through a LFM filter. The filter introduces a time lag that increased linearly with frequency at the same rate as the frequency of the transmitted up chirp signal increases. Thus, the trailing portion of the received signal takes less time to pass through and lag portion takes longer time to pass through. In this way, the pulse is compressed with a shorter width and higher amplitude. A simplified diagram of LFM procedure is shown below in Figure 5.

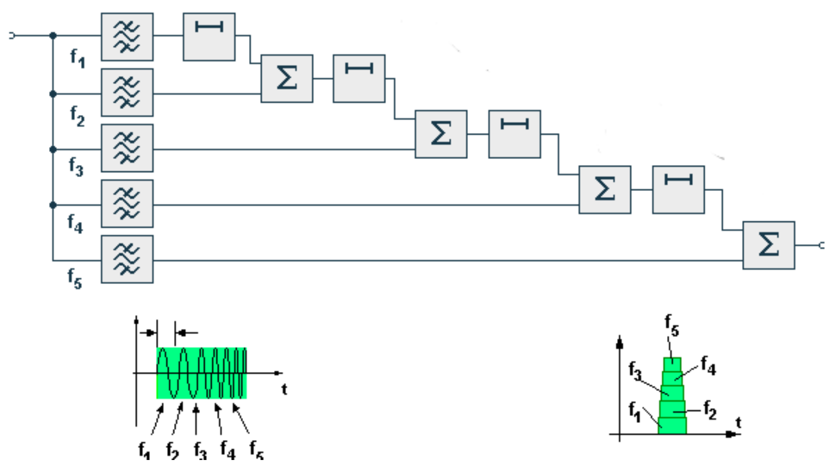


Figure 5: block diagram for LFM explanation [10]

In the binary phase modulation method, it is like a digital circuit. Only two phases are used, 0 and 180 degrees. In the modulation process, a delay line is used to match the received signal's phase exactly. A procedure of BPM is described below and in Figure 6.

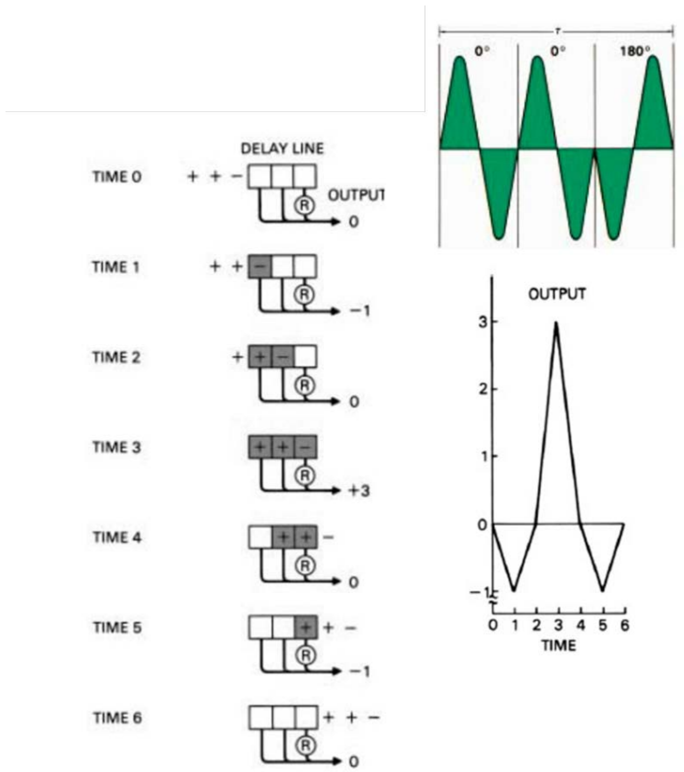


Figure 6: Procedure of BPM

“R” is the phase reversal. The phase of the signal would be reversed when passing through “r”.

The output of BPM is compressed compared with input signal.

2.6 Doppler Effect

The Doppler Effect is a shift in the frequency of a wave radiated, reflected, or received by an object in motion [6]. When the direction of motion is same at the direction of wave, the frequency is decreased. Otherwise, the frequency is increased, as shown in Figure 7.

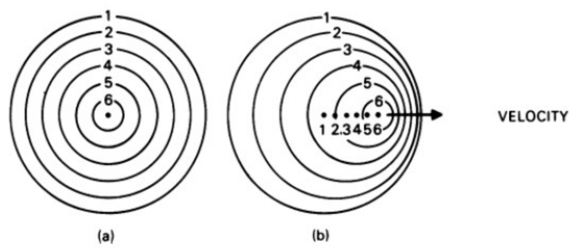


Figure 7: Wave gets compressed in the direction of motion [6]

The 11Doppler shift is caused by relative motion. We assume that relative velocity is u, wavelength for transmitted wave is f_0. Thus, Doppler frequency is:

$$f_d = 2 * u * f_0 = \frac{2 * u}{\lambda_0} \qquad (11)$$

In an airborne radar system, the Doppler Effect is a very common phenomenon because there is always as relative velocity between transmitter and transmitted signal. In our acoustic system, because the system is static for both target and receiver, the Doppler Effect does not apply.

2.7 Noise Power

When receiver detects received pulse, it receives noise at same time. The most fundamental source of noise in a system is thermal noise; however, it is also known that the noise is increased by equipment.

Assuming the noise signal is a function of time, x(t). Due to *Signals, Power and RMS,* the formula instantaneous power P(t) of our receiving signal is $P(t) = C * x^2(t)$ [10]. In our system, because C is only a proportion for the instantaneous power, we can set C constant to be 1. Thus, the instantaneous power is:

$$P(t) = x^2(t) \tag{12}$$

Thus, the average power in one period is:

$$P_{ave} = \frac{1}{T}\int_{t_0}^{t_0+T} P(t)\, dt = \frac{1}{T}\int_{t_0}^{t_0+T} x^2(t)\, dt \tag{13}$$

In LabVIEW system, it would automatically sample the signal in the receiving channel and store it into a CSV file. Thus, it sum for discrete signals should be used rather than the continuous integral for.

$$= \frac{1}{T}\sum_{T} P[n] \tag{14}$$

Due to sampling, the received signals becomes a discrete signal. T is the sampling period. We use $\frac{1}{T}$ to normalize the sum of received signals.

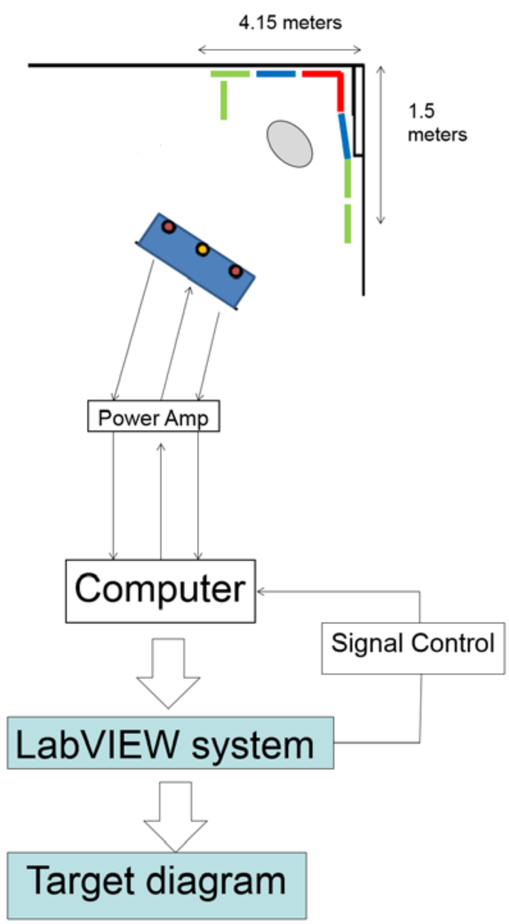
For the RMS value, it is as same as noise power, but it is more useful because RMS value could reflect the real value of power compared with echo responses. Because the instantaneous power is the square of the received signal, the average received signal should be the root-mean-square value of the average power.

$$x_{rms} = \sqrt{P_{ave}} \tag{15}$$

3. Radar Architecture and Design

3.1 System Architecture

The radar system contains two parts, hardware equipment and software in the form of the LabVIEW. The configuration of acoustic radar system is shown below in Figure 8.



Legend:

Grey circle	Iron panel, target
Yellow dot	Transmitter
Red dot	Microphones, as receivers
Colorful lines	Absorbers
Blue rectangular	Table

Figure 8: Configuration of acoustic radar system and legend

By connecting between LabVIEW and hardware, the LabVIEW system could control transmitted signal and store and process received signals. One transmitter and two receivers were placed in a line, which is orthogonal to the direction between the radar and the target, a panel. A “quiet” corner was constructed by placing foam panels on the wall (These were actually electromagnetic absorber panels, but lab based experiments demonstrated they also worked well at absorbing ultrasound radiation). We use power amplifier to amplify both transmitted and received signals.

The radar was configured to use two receivers and a single transmitter to simulate human's or bat's ears and mouth.   To minimize the cross-talk between the transmitter and receivers—signal that propagates directly from the speaker to the microphones—sound baffles were placed around the speaker.

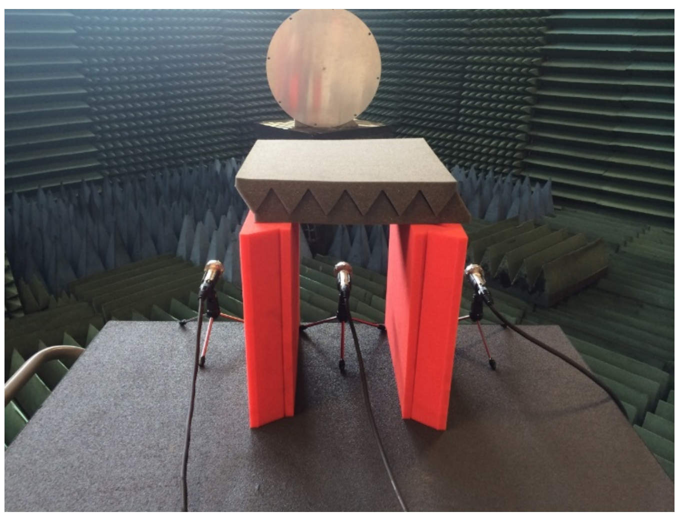


Figure 9: transmitter, receivers and panel

3.2 LabVIEW Development

The Front Panel of LabVIEW is shown in Figure 10. This Front Panel has three main parts, plotting diagram, indicator and controllers.

After development, there were five controllers to control types of transmitted pulses (I), pulse width (II), sampling rate (III) and pulse repetition frequency (IV) and transmitted pulse frequency (V) respectively.

Five new indicators were added into the Front Panel, fixed sampling rate (I) and PRF (II) were output from a new algorithm, which mentioned in section 3.2.1. Unambiguous range indicator (III) was a result for measure maximum detecting range. Two counters were set to count both samples of transmitted pulse (IV) and samples of transmitted signal (V) in one period, which is named “Pulse Width (Samples)” and “PRI (Samples)”. The rest four indicators had been there before system developed for previous research.

The third part plots the results. The first two diagrams plot the position of the target. The third diagram shows the received echo signals.

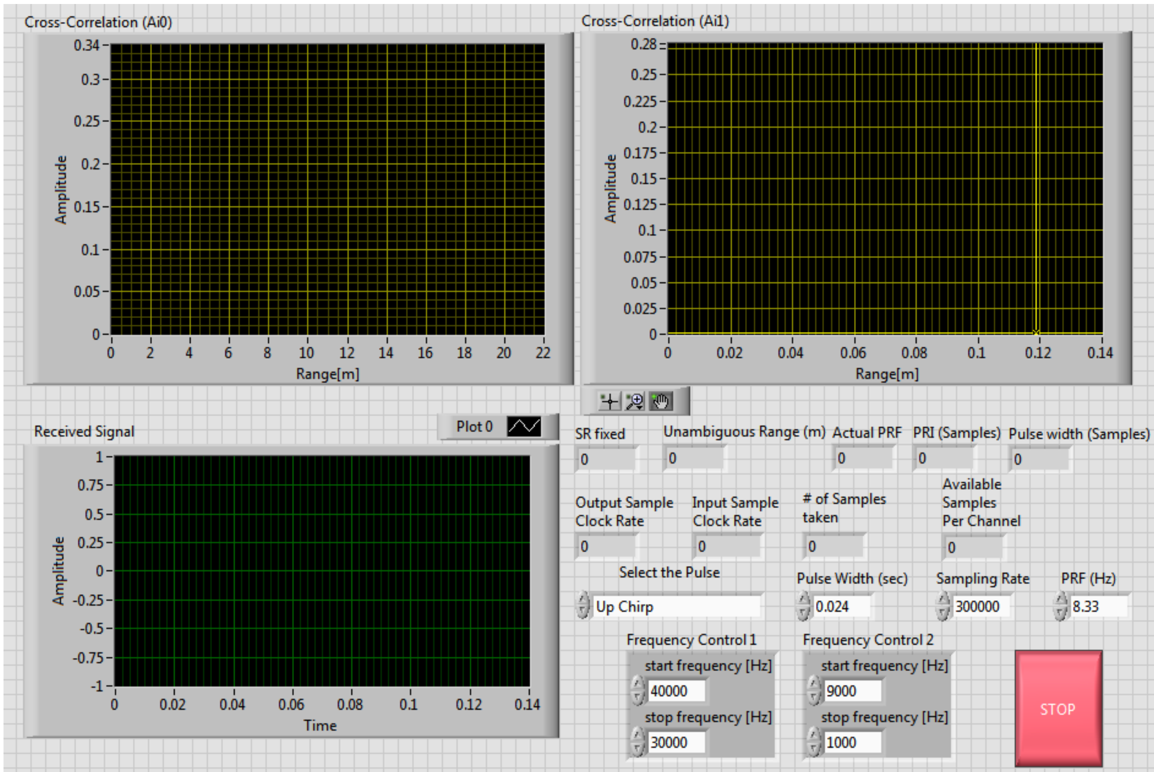


Figure 10: Front Panel of LabVIEW system

Four significant contributions were made to the system, setting up new testing waveforms, increasing stability, increased controllability and calibrating the cross-correlation output.

3.2.1 Increase Controllability

In order to do further research, the acoustic radar system needs to be more controllable and reliable, however, the previous LabVIEW system did not meet the two requirements. There were two parameters which we could control in Front Panel, sampling rate and Pulse Repetition Frequency (PRF). To conduct the further research, the pulse width and frequency controller had to be added for transmitted signals in Front Panel.

Firstly, because it is a digital system, we need to assign numbers of samples into transmitted signal. For example, in one PRI, if a total of 2400 samples were detected and filled in with signals in one period with 480 of them were assigned to transmitted pulse, thus 1960 samples needed to be padded with zeroes.

In the diagram given below (Figure 11), it shows a chirp function, connected with min and max frequency as controllers in Front Panel. An appended function was used to generate padding zeroes after the transmitted chirp function.

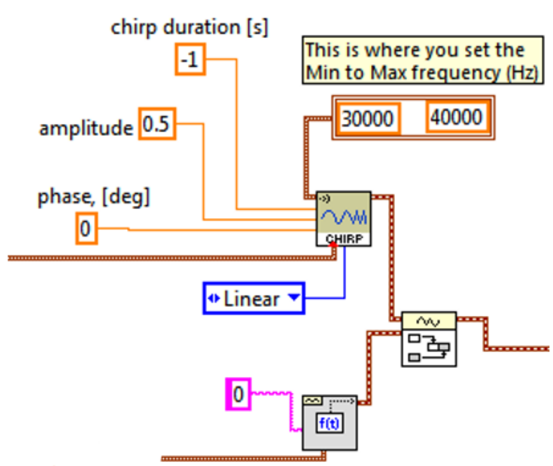


Figure 11: Padding zero after transmitted pulse

Next, we set the pulse width as controller in Front Panel in unit of seconds. The number of pulse width samples can be calculated from:

$$PW(samp) = PW(sec) * SR(fixed) \quad (16)$$

Where SR is sampling rate and PW is pulse width.

Because number of samples is an integer, thus total samples in one PRI:

$$PRI(samp) = [\frac{SR(fixed)}{PRF}] \quad (17)$$

Thus, the actual PRF that we use for transmitted signal changed correspondingly,

$$Actual\ PRF = \frac{SR\_fixed}{PRF(samp)} \quad (18)$$

After calibration, Actual PRF is the PRF we input to the system.

An indicator was created for maximum unambiguous range. As mentioned in section 2.3, the equation is given below:

$$R_{Max} = \frac{v}{2 * Actual\_PRf} \quad (19)$$

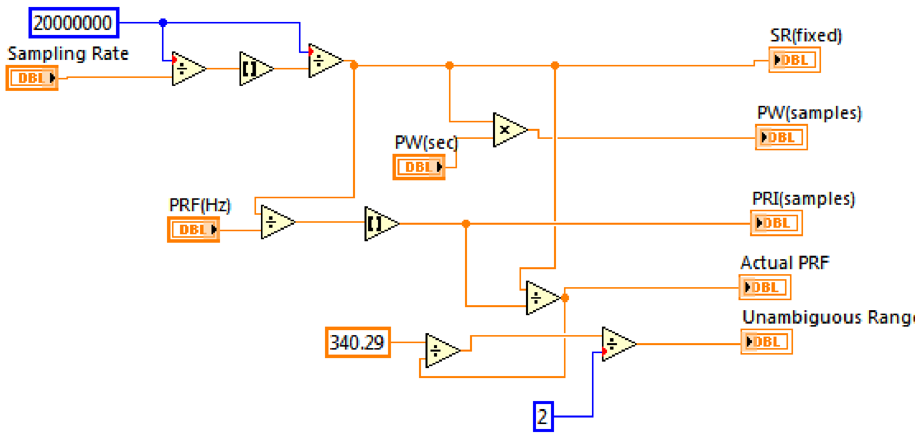


Figure 12: Stability algorithm and parameters calculation

3.2.2 New testing waveforms

Before doing the system calibration, the system had two testing signals, up chirp and down chirp.

A new testing signal was added, and up-down chirp.

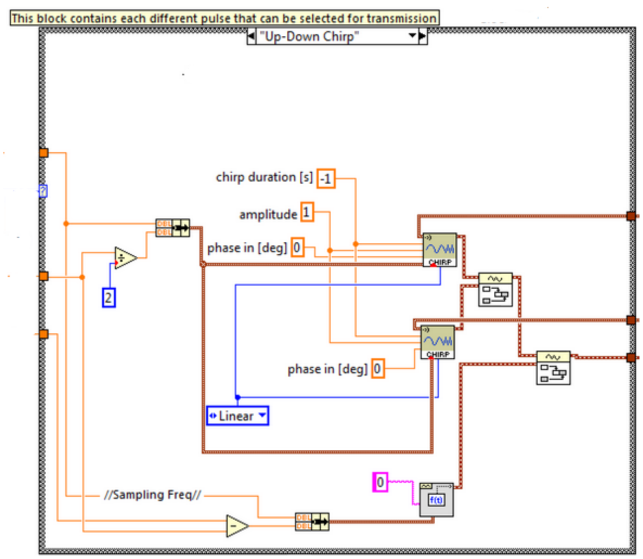


Figure 13: block for up-down chirp

3.2.3 Increase stability

The first problem was that the system was not stable. When the sampling frequency of the system was set to certain values, the system output appeared to “drift”. That is to say that the peak in the cross-correlation output that indicated a target moved from pulse to pulse, indicative of a problem with the system triggering.

In the National Instrument data acquisition (NI-DAQ) process, analog signals from real world would be converted to digital signals. Data acquisition is a procedure which sampling analog signal. The value of sampling frequency could be set in Front Panel, as shown in 3.2.1 section.

In NI-DAQ process, an actual input sampling frequency could be changed by the system automatically by its built-in function [11]; however, the input sampling frequency was still used for transmitted signals. The fixed sampling frequency was used to sample received signals. In the cross-correlation process, because of mismatch between received signal and transmitted signal, the peak would be drifted. In LabVIEW system, input values would be tested and changed by an inner function when we apply input values to NI-DAQmx.

For example, assume input sampling rate is SR. The testing procedure is given below:

$$Y.Z = \frac{2 * 10^7}{SR} \ (20)$$

Y is integer and z is decimal remainder.

$$SR_{fixed} = \frac{2 * 10^7}{Y} \ (21)$$

NI-DAQmx will change SR to SR_fixed automatically into system. Because of difference between SR and SR_fixed, peak would shift in the cross-correlation diagram.

Figure 14: algorithm in Block Diagram

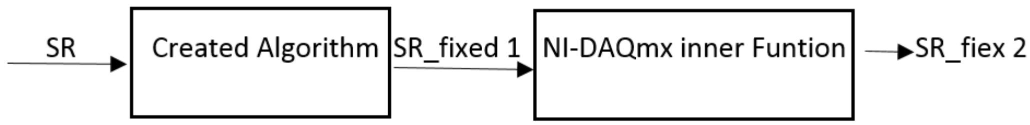


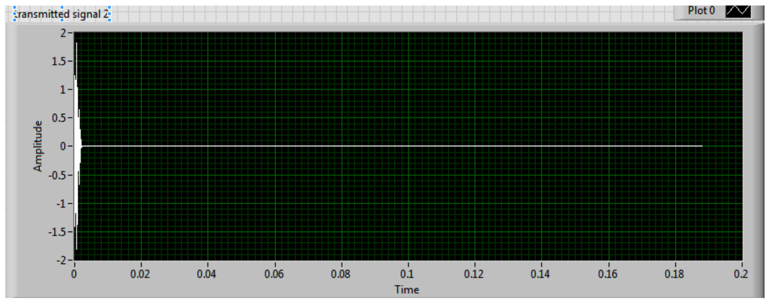
Figure 15: Simplified algorithm cascade diagram

In this solution, an algorithm was created which is as same as the inner function before it goes through the inner function. Thus, Z would be eliminate in created algorithm. Then, it could make sure that when input value goes through the inner function that Z (decimal remainder) would be zero. In other words, in the inner function, SR_fixed2=SR_fixed1, which the peak shifting would be eliminated.

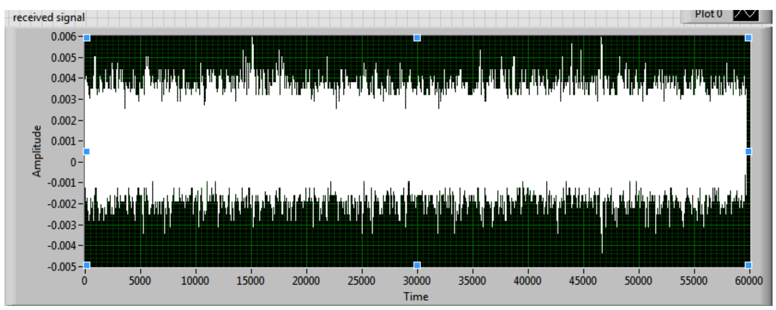
3.2.4 Calibration Cross-Correlation Diagram

The two diagram in Front Panel shows that results of cross-correlation between transmitted signals and received signals. In signal processing, cross-correlation is used to calculate time delay, and hence range to the target [12].

For example, transmitted signal is:



Received signal is:



The graph of cross-correlation is (x axis is in$10^{-2}m$):

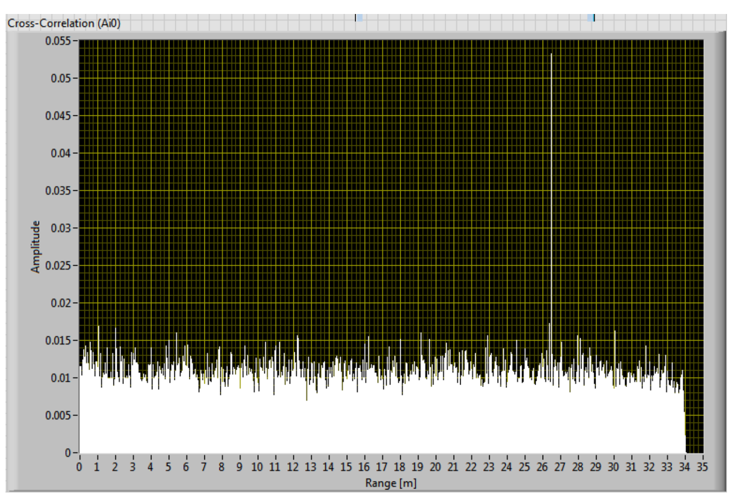


Figure 16: received, transmitted and cross-correlation waveform

The basic idea for cross-correlation in signal processing is to compare the similarity between received and transmitted signals. The peak occurs when the similarity between the received signal and the transmitted pulse is highest.

In LabVIEW, the x axis of the cross-correlation results could be either x axis of received signal or x axis of transmitted signal, which depend on the order of input.

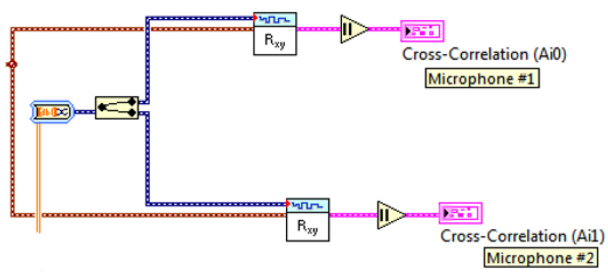


Figure 17: two channels cross-correlation

The red channel is the transmitted signal and the blue channel is the received signal. Previously, the received signal input in the first channel and transmitted signal is input in the second channel. The cross-correlation function would automatically apply the x axis of first channel to produce the result. Thus, the x axis's was sample dose in the previous system. As switching the two channels, the x axis was in one PRI. Multiplying half velocity, the x axis becomes range, as shown in Figure 16.

4. System Behavior Testing

After refining the design of the acoustic radar system a series of calibration and evaluation experiments were undertaken. The four tests are, noise power testing (I), range testing (II), pulse width testing (III) and maximum unambiguous range testing (IV).

4.1 Noise power testing

From this experiment, the main purpose is that it should be known whether there was noise coming from the background and if there was noise coming from the digital system (computer or cables lined with computer). If there was, then we should know the amplitude of the noise power because when the amplitude of noise power is comparable to the amplitude of transmitting pulse, then the echo responses would be covered by noise. Before conducting further research, system noise is the first thing that needs to be known. In this experiment, we try to figure out where the noise comes from and what is the noise amplitude. If the amplitude was too two high compared with our received signals, then some further calibration needs to be done in our acoustic radar system.

4.1.1 Experiment Methodology

Signal-to-Noise ratio is one of the most important parameter to evaluate a radar system. Because several future research would be based on this system, system noise need to be tested and determined whether in an acceptable range.

In noise power testing experiment, the testing was conducted under three situations; turn off power amplifier and turn on microphones (receivers on both sides) (I), turn on power amplifier

(II) and turn off microphones and turn off both amplifier and microphones (III). Power amplifier is a type of audio amplifier which could amplify the signals that we received.

Before testing, an assumption was that the received signal contains two parts. The first part is the echo coming from panel, and the second part could be the background reflection. In this test, the amplifier was shut down under testing; instead that pulse was set to 0. When the amplifier did not working anymore, then there were no amplified signals transmitted out of amplifier. In other words, there is supposed no echo responses from panel. Thus, the received signal shown in the Figure 19 is only outside background noise. When we shut down power amplifier, it means there is only noise (system noise) going through LabVIEW system. When we disconnect receivers, then receivers stop working and no signals could go through those channels.

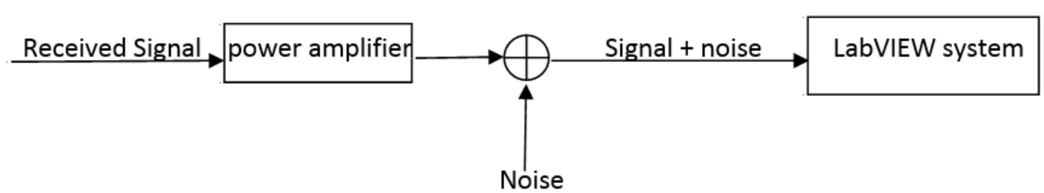


Figure 18: noise in receiving channels

4.1.2 Results & Analysis

After receiving signal, the noise power is calculated. The calculation is in 2.7 section.

i. Off power amplifier; On microphones

Channel	1st	2nd
Average power /arbitrary linearly	0.21268	0.25098
RMS-value /arbitrary linearly	0.46117	0.50098

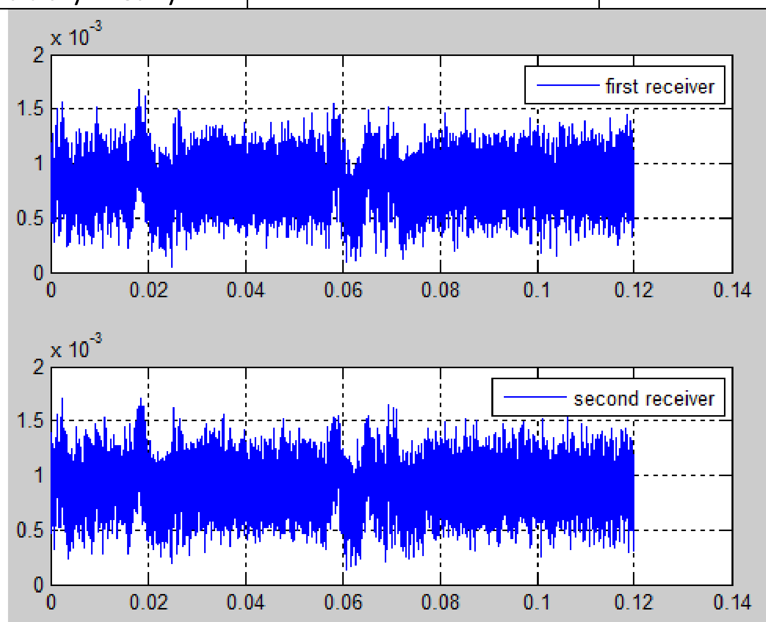


Figure 19: noise power waveform in first condition

From the first results, the highest peak around 0.0015, which was much smaller compared with regular echo responses amplitude, which is about 0.01. Thus, the noises from microphones (receivers on both sides) were acceptable for future experiments.

In addition, as marked above in the figure, two same peaks showed up in both channels at same times. As mentioned above, audio noise is a type of random noise. In this case, the two same waveforms showing in both channel at same time could be identified as real echo responses rather than random noise waveforms. As mentioned above, whether connecting or disconnecting either amplifier or receivers respectively, there was no way to avoid system noise. And system noise comes in after receiving echo responses.

ii. On power amplifier; Off microphones

Channel	1st	2nd
Average power /arbitrary linearly	0.13799	0.15936
RMS-value /arbitrary linearly	0.37147	0.39920

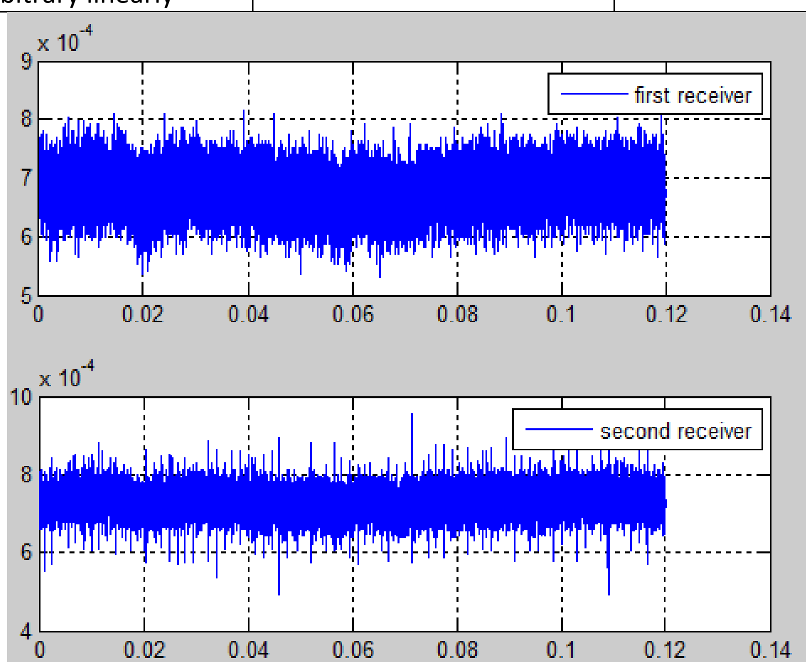


Figure 20: noise power waveform in second condition

iii. Off power amplifier; Off microphones

Channel	1st	2nd
Average power /arbitrary linearly	0.14000	0.15899
RMS-value /arbitrary linearly	0.37416	0.39873

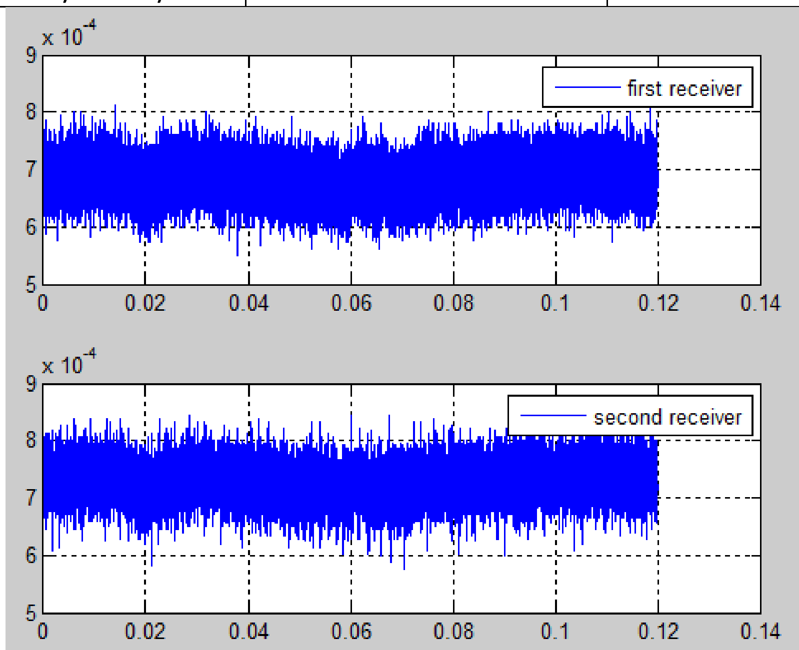


Figure 21: noise power waveform in third condition

From the third set of results, even when the amplifier was shut down and disconnected from the two receivers completely, random noise is still showing. In this case, the noise came from the hardware, such as computer, connecting cable or analog-digital converting process.

Compared with first results and third results, there was about 0.09 unit and 0.11 unit noise reduced in first and second channel respectively when disconnected microphones. It proved that even though the amplifier was turned off, the received channel were still connecting inside amplifier; however, it cannot be find that whether amplifying is working slightly or not in both channels from this comparison. The received signals are gap proximately equal in two channels.

Compared with second results and third results, two group of RMS values were extremely close, thus, turning off the amplifier has no effect on the received signals. It can be concluded that the two groups of received signals were not being amplified when going through amplifier.

4.1.3 Conclusion

From the above analysis above, there were three conclusions. Firstly, the amplitude of the noise, whether or not from outside background or system, was within an acceptable tolerance. The amplitude of received echo responses were around 0.01 as measured several times before; on the other hand, the highest peak for random noise was about 0.0015 with both system noise and outside background noise. Secondly, when the amplifier was shut down, the received signals were still able to go through the amplifier and go into the LabVIEW system. Thirdly, when turning off the microphones and the amplifier, noise was still in the system. Thus some noise came from the system which was larger than that coming from the background.

4.2 Range Testing

Radar is a system for range detection. In the acoustic radar system, acoustic wave was used to detect range between system and target. The ranging detection was accomplished by measuring time the between transmitting a pulse and receiving its echo response, which is called the timing-lag, t. However, in a real experiment, t is calculated after cross-correlation of the received and transmitted signals. Given a constant acoustic signal velocity, the detected range could be determined by the formula:

$$R = \frac{t*v}{2} \text{ [6]} \tag{22}$$

In the range testing experiment, it needs to be proved that the real range between target and our radar system is equal to measured range by LabVIEW. The real range is measure by tape, the lagging time is calculated by cross-correlation in LabVIEW system. Before doing more testing, range testing had to be finished first because some further testing could only be undertaken when the acoustic radar system could measure correctly.

The objective of this experiment is to test whether the given formula could detect the real range exactly. Because the range detection equation (22) contains two factor, and velocity of acoustic signal, it should be determined that which factor cause error if the detected range was not equal to real range.

4.2.1 Experimental Methodology

In the testing, the target was moved into different positions, but in the direction which perpendicular to the line of receivers and transmitter.

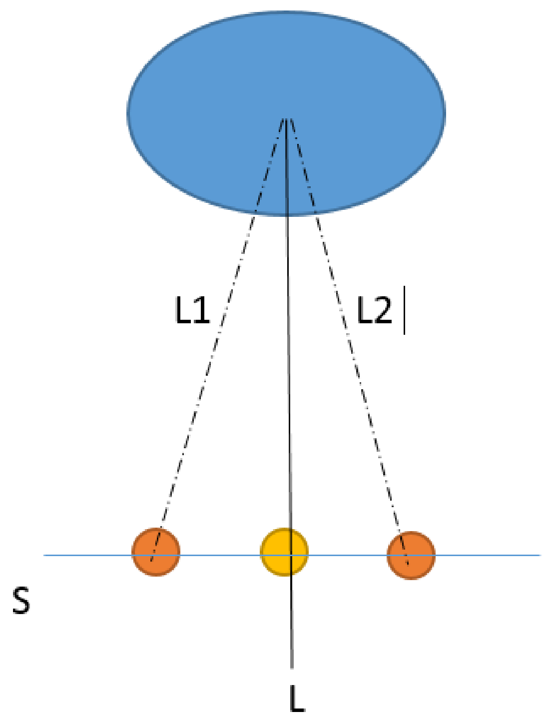


Figure 22: Panel position

As shown in Figure 22, panel was placed in different positions along the line L, and the distance from the center of the panel to the first receiver (left orange dot) and the second receiver (right orange dot) is fixed and equaled to each other. The vertical line was determined by the transmitter (middle yellow dot) and the center of panel (center of blue circle). On the other hand, the error could not be eliminated in real testing and it was difficult to set the lines S and L perpendicular to each other. Thus, to fix the problem, L1 and L2 need to be independent and that was calculated to the two detection ranges respectively. Thus, we just need to measure of relative position for the two channels independently.

In our LabVIEW system, we have series of blocks which could do cross-correlation as shown below.

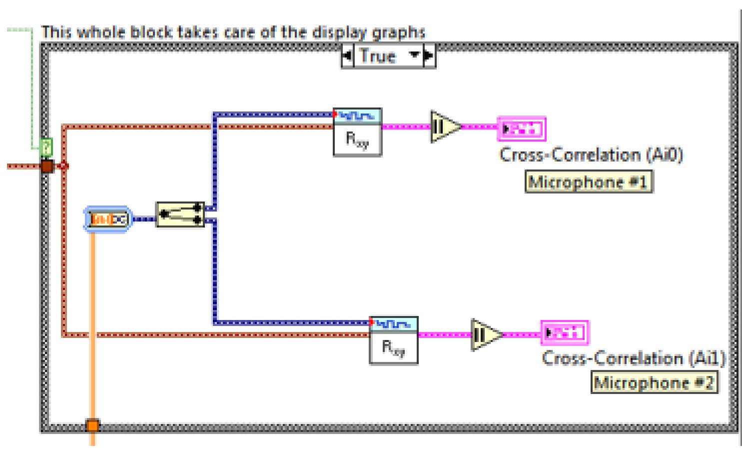


Figure 23: cross-correlation in block diagram

In the diagram, the red line is the transmitted signal. The orange line contains the two dimensional data. We separate the two dimensional data into two channels to do the cross-correlation calculation. In the signal processing, cross-correlation is a method of detection of time-lag between two signals. The results of cross-correlation shows the time-lag. By multiplying the velocity of the acoustic signals, the range-lag can be calculated, which provide the target positions.

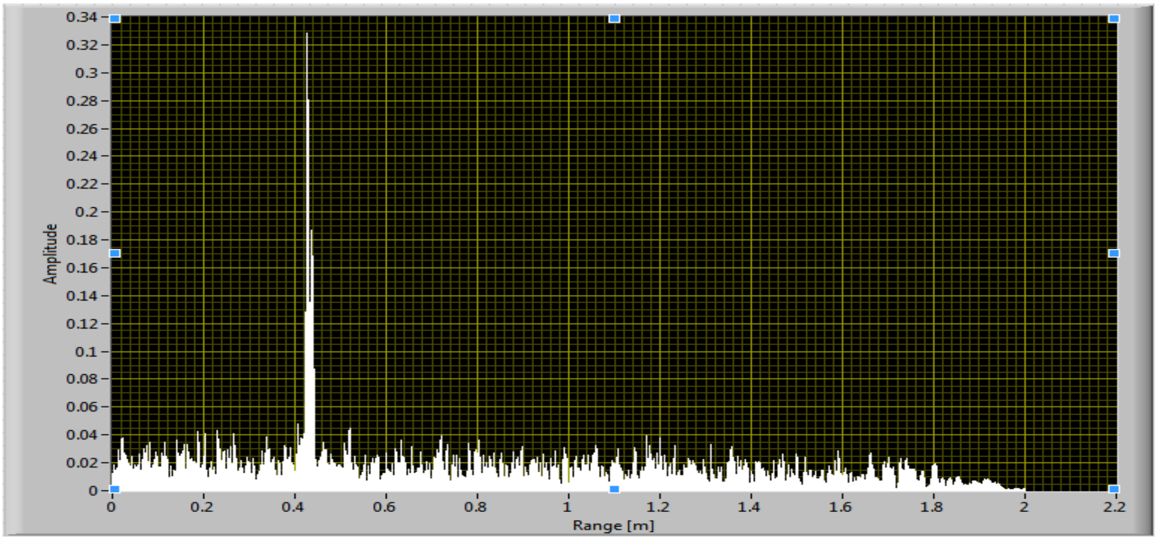


Figure 24: Detected range in LabVIEW front panel

The detected range by calculation inside the LabVIEW is shown above (Figure 24). The peak stands for the target, and the detected range is shows on the x axis. However, the digits are not enough to show detected range. Thus, we use MATLAB to measure detected range by plotting the data in CSV file.

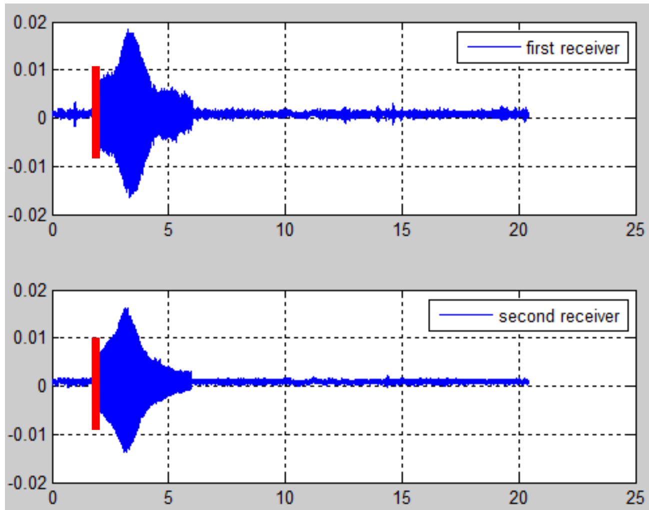


Figure 25: detected range by MATLAB plot

An example of a MATLAB plot is shown in Figure 25. Target is marked by the red line. X axis is range in meters, the y axis is the amplitude of the received signals. The position of the vertical line of the waveform along the x axis is the target position, as marked in Figure 25. The waveform is an echo response coming back from the target. In this case, the transmitting signal is up chirp, with a frequency changing from 30000 to 40000 Hz linearly.

4.2.2 Results & Analysis

By conducting the experiment, the target (blue panel) was placed at different positions and the ranges were measured using both method. The results are shown below.

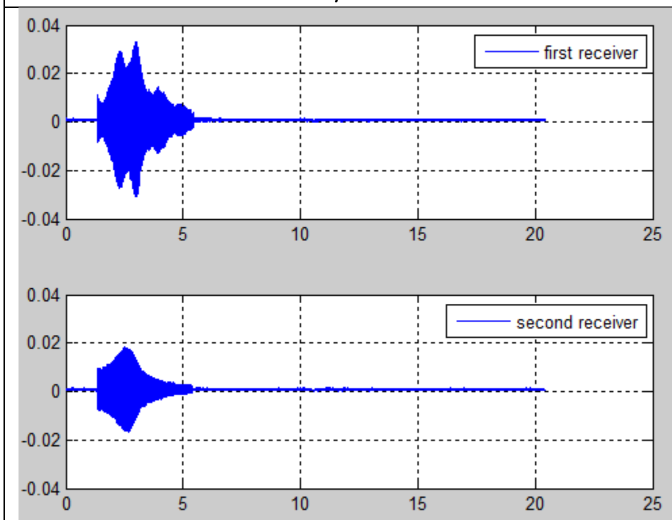
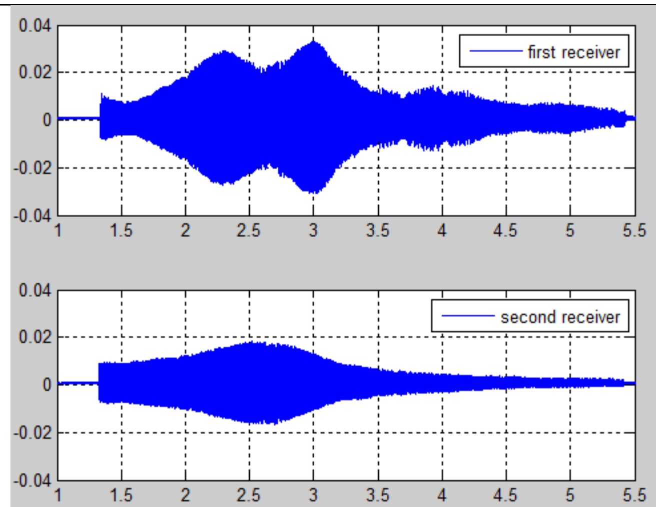
Ranges from left channel to target (1st channel)/cm Received signal from both channel	Ranges from right channel to target (2nd channel)/cm Zoom-in figures for both channel
135/141	137/141.1
	

Table 1: An example result for measured ranges

The waveforms that showing in Table 1. As moving further and further away from our radar system, the amplitude of received signals got lower and lower. It means that the received signal got weaker when we move target further, because it was more attenuated for acoustic signals than electromagnetic waves propagating through air. Moreover, by zooming in, echo responses had higher amplitude in the first channel.

Then, data was placed into Figure 26. The x axis is real range and y axis is detected range.

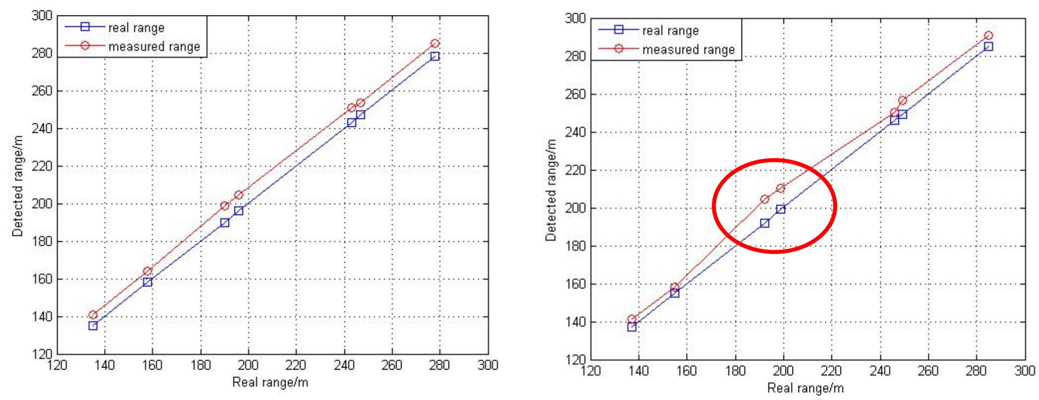


Figure 26: Real range and detected range in both channels

The blue line is real range between target and acoustic radar system. Because the blue point is measured by tapes, the values of blue points in y axis are equal to their values in x axis. In other words, the angle between blue line x axis is 45 degree. The blue line is a “standard” line; we use it to compare with detected range. The values for red points in y axis are the detected range and the values for red points in x axis are real range. Red point reflect that the detected range corresponding to the real range. By observing the range-range diagram in both channels, there is a consistent delay in both channels. Furthermore, as marked in second channel, the delay for the real range from 160 to 240 cm, the delay becomes higher than consistent delay in first channel.

4.2.3 Conclusion

In the range testing, it can be concluded that there were consistent delay in both channel; however, the delay in the second channel became higher in some particular position. On the other hand, a further research about the higher delay could be conducted because currently there are not enough data to display how it changes specifically.

As moving target further, received signals became lower. An acoustical wave reflection and refraction could be a reasonable explanation for the phenomenon. Additionally, the amplitude of received signals was higher in the first channel than it in the second channel. There could be two possible reasons. First, a higher power for transmitted signals in first channel could be a reasonable explanation. Second, the angle for the second receiver was not exactly facing on the center of panel.

4.3 Pulse Width Testing

In this testing, a measurement of received pulse width was undertaken. This testing signal was up chirp. Up chirp is a signal whose frequency increased with time. By testing the pulse width of received pulse, the difference between pre-set value and received pulse could be calculated.

4.3.1 Experiment Methodology

By storing received signals into CSV file, received signal was plotted by MATLAB. In the plot of MATLAB, it could be measured of the position of both tale and head of received pulses by cursor. The pulse width is the difference between two values in x axis.

Assume that the position of head is(x_1, y_1), and tale is(x_2, y_2). Because our transmitted pulse is up chirp, we have a relation:$x_2 > x_1$.

Thus, we can calculate Pulse Width:

$$\text{Pulse Width} = x_2 - x_1$$

The waveform and waveform given below:

Pre-set Pulse Width in LabVIEW system /sec	Pulse Width in Channel 1/sec	Pulse Width in Channel 2/sec
0.005	0.00498	0.00497
0.01	0.01001	0.00993
0.024	0.02399	0.02397
0.05	0.04998	0.04997
0.08	0.08001	0.07994
0.1	0.10003	0.10000

Table 2: Pulse Width comparison

4.3.2 Results & Analysis

In this part, results show that it is only different in fifth digit, when changing pulse width in third digit.

4.3.3 Conclusion

The error between measured pulse width and pre-set pulse width changes from -0.7% to + 0.0125%. From the results, one conclusion can be made based on the analysis and results. The absolute value of maximum error in percentage is 0.7%, which is in acceptable tolerance for doing further research based on the same system.

4.4 Maximum Unambiguous Range Testing

As Maximum Unambiguous Range theory mentioned above, another experiment was designed to prove the theory. Due to the Maximum Unambiguous Range theory, we know the equations:

$$R = \frac{v*(i*PRI+t)}{2} \text{ , i=0, 1, 2, 3...} \tag{23}$$

$$R_{Max} = \frac{v*PRI}{2} = \frac{v}{2*PRf}, \tag{24}$$

Among them, R is detected range, when the target is in unambiguous range, I is 0. v is the velocity of the acoustic signal. It is about 340.30m/sec in room temperature. i is a integer when the distance beyond maximum unambiguous range, R_{Max}. t is the detected responding time identified by our system. t is measure between transmitting a pulse and received an echo response.

In this testing experiment, the given equation (23) need to be proved.

4.4.1 Experiment Methodology

To increase the distance between target and acoustic radar system, acoustic radar was moved further. In this experiment, target was placed about 250 cm away. The range from target to first receiver was 247 cm and to second receiver was 249.2 cm. Next, we set Pulse Repetition Frequency in LabVIEW front panel to 85. The PRF makes Maximum Unambiguous Range to 200 cm by given equation (24), which satisfied to experiment condition.

4.4.2 Results & Analysis

By running acoustic radar system, the peak diagram shows in Front Panel.

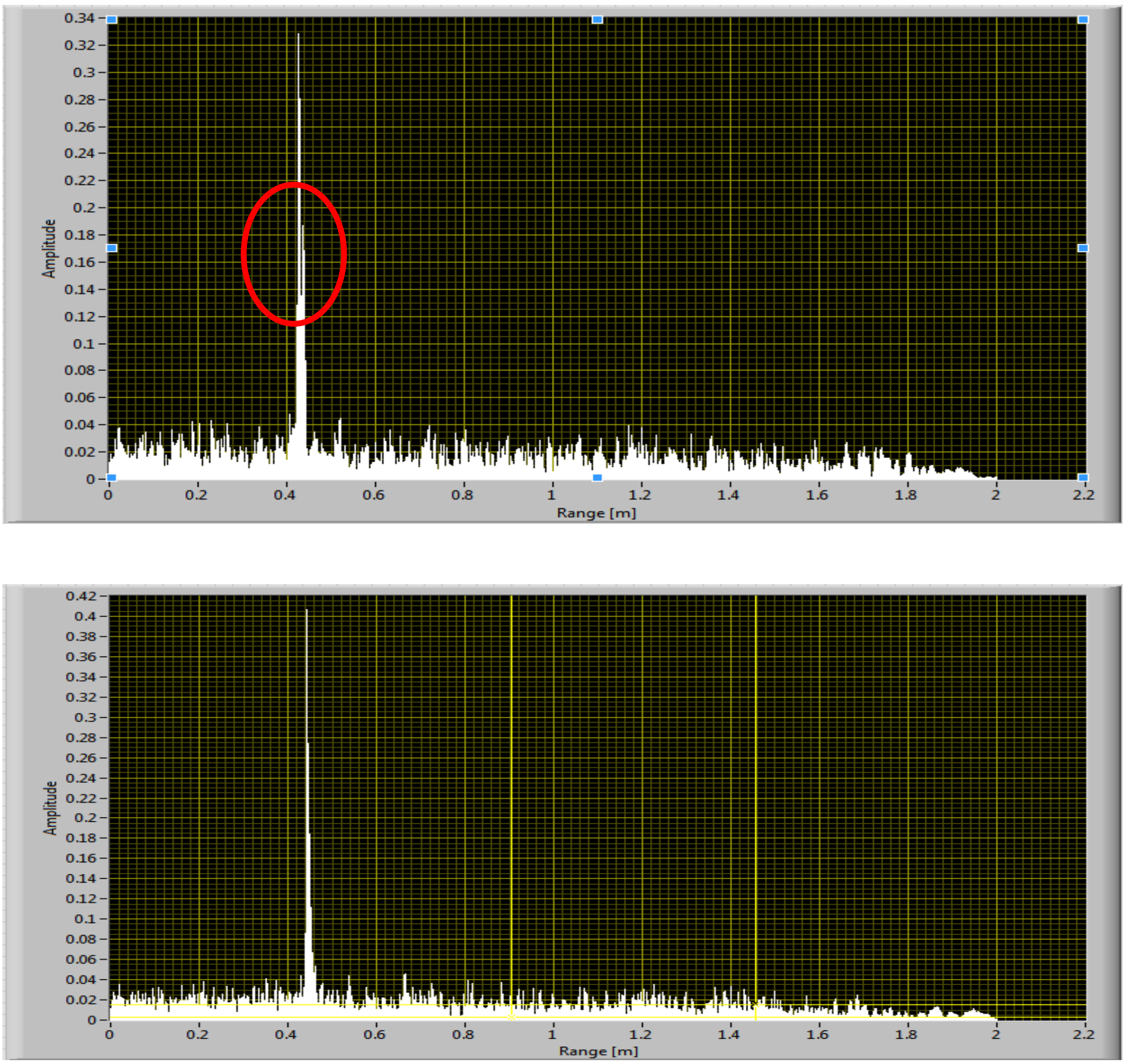


Figure 27: Peak diagrams in LabVIEW Front Panel for channel 1 and 2

Then, the waveforms for both channels was plotted by MATLAB and measured by cursor.

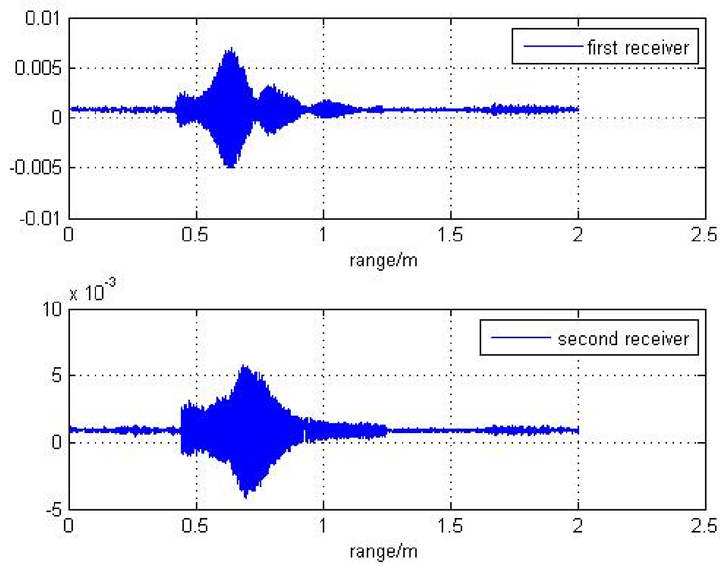


Figure 28: Waveforms from both channels

By Cursor measure measurement, we know that the detected range for channel 1 and 2 are, 50.81 and 52.6 cm respectively.

From (23),

$$R = \frac{v * (i * PRI + t)}{2}$$

$$= \frac{v * i * PRI}{2} + \frac{v * t}{2}$$

$$= i * R_{Max} + R_{Det} \tag{25}$$

In this experiment, i is equal to 1 because

$$R_{Det} < R_{real} \approx 1 * R_{Max} + R_{Det} \quad (26)$$

The difference between two real range and sum of maximum unambiguous range and detected range is

$$D = (R_{Max} + R_{Det}) - R_{real} \quad (27)$$

Then, we summary two groups of data in two one table:

Channel	Real Range/cm	Detected Range/cm	Maximum Unambiguous Range/cm	Difference
1	247.00	48.81	200.00	1.81
2	249.20	50.60	200.00	1.40

Table 3: Experiment Data for both channels

As we mentioned in Range Testing part, there was a slightly difference between detected range and real range. D was caused by system delay.

Moreover, in the peak diagram, there was a peak with a sub-peak showing in first channel, as marked above. By plotting waveform in first channel, the body of waveform had a trough, which is corresponding to sub-peak.

4.4.3 Conclusion

In this experiment, we prove the maximum unambiguous range theory. The results were not exactly as same as the equation (24). The system delay make the ($R_{Max} + R_{Det}$) larger than R_{Real}.

From the sub-peak, any variation in received waveform would connect with the peak value in peak diagram. Additionally, it was correct to measure target range by MATLAB plot rather than cursor in peak diagram, because the location of the peak is not the beginning (head) of waveform. The peak in the diagram is corresponding to the peak of received waveform.

5. Summary & Conclusions

In the experiment, the research mainly contains two main parts, system calibration and behavior testing.

In the first part, we finished both hardware architecture and LabVIEW system calibration. In the hardware architecture, we set up a second channel for receiver with absorbers to separate the transmitter and two receivers. When we have two receivers, our system can receive signals in two channels which could simulate human's and bat's echolocation ability. Then, we increase controllability and stability of LabVIEW system. We add both controllers and indicators in Front Panel. In addition, we create algorithm in Block Diagram to avoid peak shifting. In order to conduct further research, we add one more testing signal, up-down chirp. We had down chirp and up chirp before, however, it is necessary to apply the new testing signal into system because of different properties of those signals.

In the second part, we have four experiments to test our system behavior. In the first testing, we test noise because noise is the most important for every system. To find out where noise come from, we test system in three situations. From the results and analysis, we conclude that there are two parts of noise. One is from background, the other is from system. Although two channel have slightly different noise, both of them are in acceptable tolerance. In the second experiment, we test range-detection ability because the main purpose for every radar system is to measure the range between target and radar. We conclude that there is a consistent system delay for both channel. In other word, the detected range is slightly longer than real range in both channels. In addition, we notice that the delay in second channel is longer when we move target to some specific position. In third experiment, we test pulse width. Because our acoustic radar system is a pulsed radar system, pulse width is a very important parameter. By comparing measured pulse width and pre-set pulse width, we conclude that the difference is only from -0.7% to + 0.0125% which is in acceptable tolerance. Based on the results, we think that transmitting procedure could compress pulse width. In the fourth experiment, we use acoustic radar to prove Maximum Unambiguous Range theory. In this research, we find out more details about acoustic radar. The testing result shows that the two receivers and the transmitter are qualified to be conducted into more research.

Our ultimate goal is to help the man-made system based on the further research, and there is no need for standards and regulatory consideration.

5.1 Recommendations

We have three plans in future research. First, because our ultimate goal is to study echolocation for blind human. Our next research is comparison waveforms by transmitting bats' and human's signals. Secondly, in other to have a better SNR, we could design a matched filter to eliminate noise. Finally, we noticed that transmitted pulse and received pulse are not in a same shape. By control system, we could look at the procedure traveling from transmitter to receiver as a system.

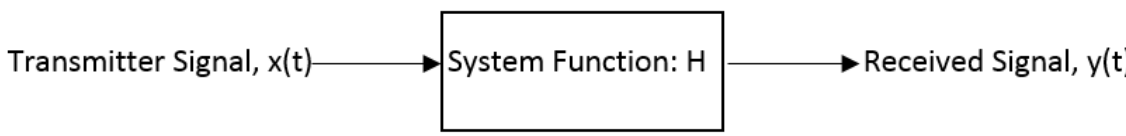


Figure 29: system function

In order to calculate system function, we can apply the formula:

$$H(w) = \frac{fft(y)}{fft(x)} \tag{28}$$

Once we know system function, H, we could design a filter which has high efficiency for background noise elimination.

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Appendix 1

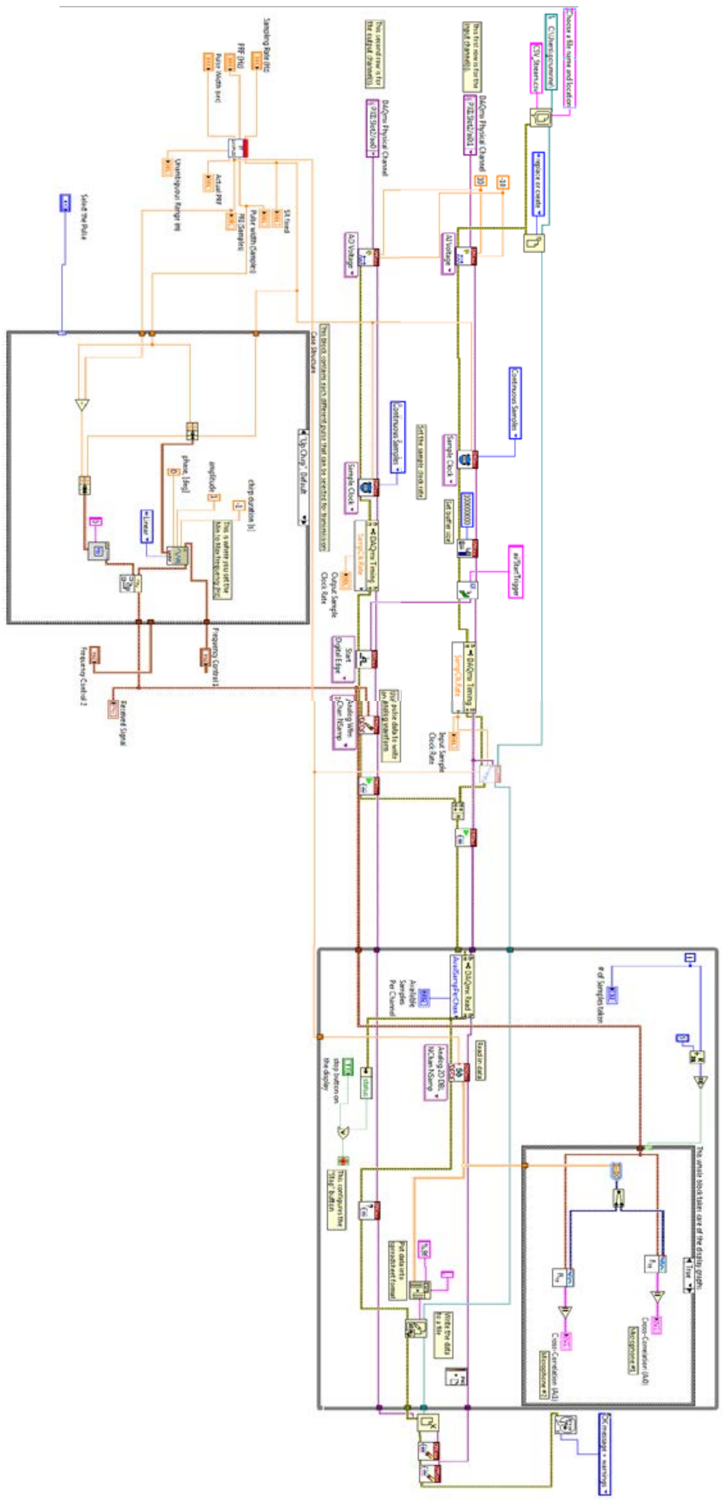

Figure: Block Diagram of LabVIEW system

Appendix 2

```
%% main fuction
clc
clear all
close all
fileName='CSV_Stream.csv';
% read file name
[lineIndex_nontitle]=read_mixed_CSV(fileName);
%lineIndex_nontitle is how many rows which contaning data
[fir_rec,sec_rec,row_num]=algorithm_rec(lineIndex_nontitle,fileName);
%fir_rec is fixed firt channel data
%sec_rec is fixed second channel data
%row_num is how many numbers for one row.
%% X axis Calculation (time)
Fsamp=csvread(fileName,2,1, [2,1,2,1]); %read sampling rate from CSV
file.
x=0:row_num-1;
range_x=170.15.* x/Fsamp; %apply x into time
%% background noise subtraction
% fileName_noise='CSV_Stream_noise.csv';
% [lineIndex_nontitle_noise]=read_mixed_CSV(fileName_noise);
%
[fir_rec_noise,sec_rec_noise,row_num_noise]=algorithm_rec(lineIndex_non
title_noise,fileName_noise);
% fir_rec_new=fir_rec;
% sec_rec_new=sec_rec;
% range_freq=range_x.*(Fsamp/row_num);
% fft_fir_rec=fft(fir_rec);
% fft_sec_rec=fft(sec_rec);
%% plot
figure;
subplot(2,1,1)
plot(range_x,fir_rec)
% axis([0 25 -0.01 0.01])
legend('first receiver')
grid on
subplot(2,1,2)
plot(range_x,sec_rec)
% axis([0 25 -0.01 0.01])
legend('second receiver')
grid on

% figure;
% subplot(2,1,1)
% plot(range_x,fir_rec)
% axis([2.5 7 -0.01 0.01])
% legend('first receiver')
```

```
% grid on
% subplot(2,1,2)
% plot(range_x,sec_rec)
% axis([2.5 7 -0.01 0.01])
% legend('second receiver')
% grid on

% figure;
% subplot(2,1,1)
% plot(range_freq,fft_fir_rec)
% legend('first frequency plot')
% subplot(2,1,2)
% plot(range_freq,fft_sec_rec)
% legend('second frequency plot')
disp('program running successfully!')
```

Appendix 3

```
%% Calculation length of column
function [lineIndex_nontitle]=read_mixed_CSV(fileName)
fid = fopen(fileName,'r');   % Open the file
  lineArray = cell(100000,1);     % Preallocate a cell array (ideally
slightly                                %#   larger than is needed)
  lineIndex = 1;               % Index of cell to place the next line
in
  nextLine = fgetl(fid);       % Read the first line from the file

  while ~isequal(nextLine,-1)         % Loop while not at the end of
the file
    lineArray{lineIndex} = nextLine;  % Add the line to the cell array
    lineIndex = lineIndex+1;          % Increment the line index
    nextLine = fgetl(fid);            % Read the next line from the
file
  end
  lineIndex_nontitle=lineIndex-8;
end
%this part of function is to read column number
%the return value of 'lineIndex_nontitle'not including first 8
'useless' column,
```

Appendix 4

```
function
[fir_rec,sec_rec,row_num]=algorithm_rec(lineIndex_nontitle,fileName)
% in this function, we need to design a alogrithm for both channels. We
% could simply use average number, or, we could sum absolute value then
% average; or we could sum square... in this case,i just use average.
%two return vectors are amplitude for two channels respectively.
%% Average for receiver 1
% in this part function, we need to calculate average all values for
each column
% another thing we need to notice is that there are two receivers,
% so odd column is for first receiver, even column is for second
receiver
  data=csvread(fileName,9,0); % import data into a matrix
  row_num=length(data(1,:));
  fir_rec_sum=zeros(1,row_num);  % '40000' is changed with sampling
rate
  sec_rec_sum=zeros(1,row_num);

  %% loop for first receiver
  for lineIndex=1:2:lineIndex_nontitle-3        % Index of cell to
place the next line in
      nextLine=data(lineIndex,:);
      fir_rec_sum=fir_rec_sum+nextLine;
  end
  %% loop for second receiver
  for lineIndex=2:2:lineIndex_nontitle-2       % Index of cell to place
the next line in
      nextLine=data(lineIndex,:);
      sec_rec_sum=sec_rec_sum+nextLine;
  end
%% average calculation
% we read total column number in previous before, we need to calculate
how
% many number for two receivers respectively
column_rec=lineIndex_nontitle/2;
fir_rec=fir_rec_sum./column_rec;
sec_rec=sec_rec_sum./column_rec;
end
```

Appendix 5

```
%% main fuction for noise power
clear all
clc
fileName='CSV_Stream_2.csv';
% read file name
[lineIndex_nontitle]=read_mixed_CSV(fileName);
%lineIndex_nontitle is how many rows which contaning data
[fir_rec,sec_rec,row_num]=algorithm_rec(lineIndex_nontitle,fileName);
%fir_rec is fixed firt channel data
%sec_rec is fixed second channel data
%row_num is how many numbers for one row.
%% X axis Calculation (time)
Fsamp=csvread(fileName,2,1, [2,1,2,1]); %read sampling rate from CSV
file.
time_x= row_num/Fsamp; %apply x into time
%% average power
p_1=0;
p_2=0; % average power for two channels
 for k=1:1:row_num
     p_1=p_1+fir_rec(k)^2;
     p_2=p_2+sec_rec(k)^2;
 end
 p_1=p_1/time_x;
 p_2=p_2/time_x;
 x1_rms=sqrt(p_1);
 x2_rms=sqrt(p_2);% x_rms is rms value for p
 description=['1st average power ','2nd average power ','1st RMS ','2nd
RMS'];
 value=[num2str(p_1),' ',num2str(p_2),' ',num2str(x1_rms),'
',num2str(x2_rms)];
 disp(description);
 disp(value)
```